

REPORT DOCUMENTATION PAGE		READ INSTRUCTIONS BEFORE COMPLETING FORM
1. REPORT NUMBER ESD-TR- 86-043	2. GOVT ACCESSION NO.	3. RECIPIENT'S CATALOG NUMBER
4. TITLE (and Subtitle) Wideband Chirp-Transform Adaptive Filter		5. TYPE OF REPORT & PERIOD COVERED Journal Article
		6. PERFORMING ORG. REPORT NUMBER MS-6957
7. AUTHOR(s) Arsenault, Duane R. Dr.,		8. CONTRACT OR GRANT NUMBER(s) F19628-80-C-0002
9. PERFORMING ORGANIZATION NAME AND ADDRESS Lincoln Laboratory, M.I.T. P.O. Box 73 Lexington, MA 02173		10. PROGRAM ELEMENT, PROJECT, TASK AREA & WORK UNIT NUMBERS 7X263304D215
11. CONTROLLING OFFICE NAME AND ADDRESS Office of the Chief of Research & Development Department of the Army, The Pentagon Washington, MA 20310		12. REPORT DATE 11 October 1986
		13. NUMBER OF PAGES 7
14. MONITORING AGENCY NAME & ADDRESS (if different from Controlling Office) Electronic Systems Division Hanscom Air Force Base Bedford, MA 01730		15. SECURITY CLASS. (of this report) UNCLASSIFIED
		15a. DECLASSIFICATION DOWNGRADING SCHEDULE n/a
16. DISTRIBUTION STATEMENT (of this Report) Approved for public release; distribution unlimited.		
17. DISTRIBUTION STATEMENT (of the abstract entered in Block 20, if different from Report)		
18. SUPPLEMENTARY NOTES 1985-IEEE Ultrasonics Symposium		
19. KEY WORDS (Continue on reverse side if necessary and identify by block number) Spectral monitoring, Multiple Narrowband Signals, Wideband Chirp-transform Adaptive-filter		
20. ABSTRACT (Continue on reverse side if necessary and identify by block number) <div style="display: flex; justify-content: space-between;"> <div style="width: 45%;"> <p>A wideband chirp-transform adaptive-filter system is described that is capable of providing both spectral monitoring and the excision of multiple narrowband signals. The 80-MHz-bandwidth system processes a wideband continuous 300-MHz center-frequency input signal in 10-<math>\mu</math>s segments, using two identical subsystems to achieve 100-percent duty cycle. Spectral analysis is achieved with a multiply-convolve-(multiply) configuration in the forward section of each subsystem. Spectral components can then be excised by time gating of the forward transforms. Subsequently, inverse-transform sections employing a (multiply)-convolve-multiply configuration recreate 12.5-<math>\mu</math>s-long segments of the filtered time-domain signal. Using a principle described previously, these output segments from the two subsystems are overlapped by 25 percent and combined coherently in order to cancel artifacts</p> </div> <div style="width: 45%;"> <p>introduced by the input segmentation. The coherence required is of the order of 7 ps between adjacent segments and is obtained by using the reflective array compressors bilaterally so that both subsystems share common devices. The system provides one 100-kHz-resolution spectrum every 10 <math>\mu</math>s at a delay of 20 <math>\mu</math>s. The overall signal delay of the adaptive filter is 40 <math>\mu</math>s. Dynamic range in excess of 30 dB has been achieved. Excision of frequency subbands as narrow as 400 kHz by up to 30 dB will be described.</p> </div> </div>		

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### ABSTRACT

A wideband chirp-transform adaptive-filter system is described that is capable of providing both spectral monitoring and the excision of multiple narrowband signals. The 80-MHz-bandwidth system processes a wideband continuous 300-MHz center-frequency input signal in 10- $\mu$ s segments, using two identical subsystems to achieve 100-percent duty cycle. Spectral analysis is achieved with a multiply-convolve-(multiply) configuration in the forward section of each subsystem. Spectral components can then be excised by time gating of the forward transforms. Subsequently, inverse-transform sections employing a (multiply)-convolve-multiply configuration recreate 12.5- $\mu$ s-long segments of the filtered time-domain signal. Using a principle described previously, these output segments from the two subsystems are overlapped by 25 percent and combined coherently in order to cancel artifacts introduced by the input segmentation. The coherence required is of the order of 7 ps between adjacent segments and is obtained by using the reflective array compressors bilaterally so that both subsystems share common devices. The system provides one 100-kHz-resolution spectrum every 10  $\mu$ s at a delay of 20  $\mu$ s. The overall signal delay of the adaptive filter is 40  $\mu$ s. Dynamic range in excess of 30 dB has been achieved. Excision of frequency subbands as narrow as 400 kHz by up to 30 dB will be described.

### I. Introduction

Real-time adaptive filtering and spectral analysis are desirable functions for inclusion in the front end of a wideband receiver. Adaptive filtering can be used to improve reception by dynamically suppressing narrowband interferers within the band. Spectral analysis can be used to provide information required to effectively bring other suppression resources into play. Both of these functions can be simultaneously provided using an appropriately configured chirp-transform system (CTS) (refs. 1 and 2). Such a system can be designed to operate on wideband signals within the VHF/UHF range using surface-acoustic-wave (SAW) chirp filters.

\*This work was supported by the Department of the Army.

A CTS adaptive filter functions by first transforming a signal into a time-mapped version of its Fourier transform. In this form the signal is suitable for time gating for spectral excision and time sampling for spectral analysis. The system then inverse transforms the time-gated transform to produce a filtered version of the original signal. This two-step transformation process delays the signal by only tens of microseconds.

A single Fourier transform for a very long signal cannot be provided without very long time delays and unrealizably large chirp filters. To circumvent this problem, a continuous signal is broken up into contiguous segments that are processed individually and then reconstructed into a continuous signal. This process is equivalent to passing a continuous signal through a linear filter so long as the segments, which are elongated by filtering, are permitted to overlap on summation. When a narrowband signal is filtered and overlap summation (ref. 3) is not used, residual signal will remain around the segment boundaries.

This paper describes the current status of an ongoing effort to develop a CTS adaptive filter providing maximal narrowband suppression of continuous signals. This development has led to a dual-channel structure utilizing a high degree of component sharing to minimize differences between channels as well as the number of system components. The system is intended for the prefiltering of a 300-MHz-center-frequency minimum-shift-key (MSK) modulated 92.5 Mchip/s PN-coded signal prior to matched-filter detection and demodulation. The goal is to provide upwards of 30 dB of narrowband suppression uniformly over all time at the CTS output. Results of a simulation are presented to indicate the maximum possible performance of the system under ideal (zero-distortion) conditions. The expected dependence of narrowband suppression on the level of amplitude and phase distortion is specified and experimental results are given.

### II. Adaptive Filtering Fundamentals

The basic principles of operation of a CTS adaptive filter can be understood by referring to Fig. 1. This figure shows a CTS structure consisting of the basic elements sufficient for the adaptive filtering of a single isolated



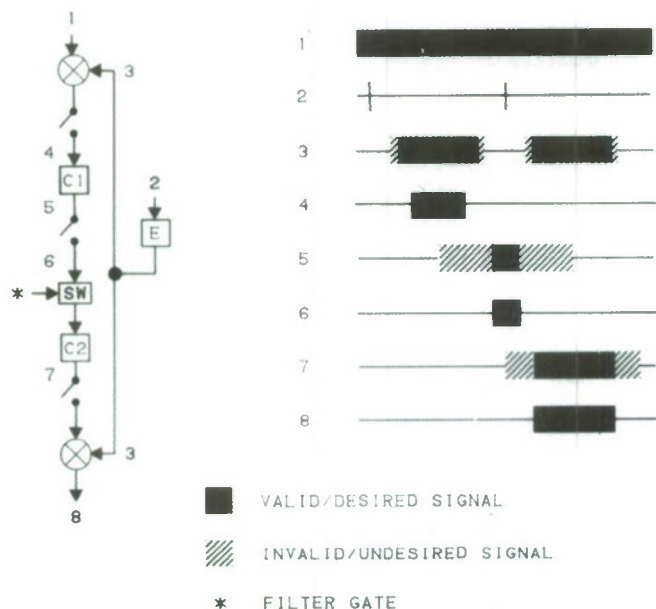


Fig. 1 Chirp-transform adaptive filter and signal timing.

segment of a continuous signal. Signal timing is also shown indicating the timing of valid and invalid signal evolving at pertinent points within the system. The intention of this figure is to draw attention to the problems associated with continuous adaptive filtering.

A segment of a continuous input signal, when chirp modulated and passed through a chirp filter, produces the chirp-modulated Fourier transform of the segment. This transform can then be time sampled for spectral analysis and time gated for narrowband excision. When this transform is subsequently passed through a second chirp filter, the chirp-modulated inverse transform of the segment is produced. Chirp demodulation then results in a filtered version of the input segment. This segment, however, is elongated by filtering and for relatively smooth spectral notches the major portion of this elongation is given approximately by the inverse of the (6-dB) notch width.

In Fig. 1 the chirps required for chirp modulation and demodulation are both generated by impulsing an 'expander' chirp filter denoted E. The impulse response of the expander is long enough to demodulate the output segment, which is longer than the input segment due to filtering. The chirp is, therefore, longer than required for input chirp modulation. The valid Fourier-transform output from the chirp filter, C1, is equal in duration to that of the impulse response of C1 minus the duration of the input segment. Input-segment-wide invalid signal precedes and succeeds this transform and is gated away prior to chirp filter C2. The valid inverse-transform output from C2 is equal in duration to that of the

impulse response of C2 minus the duration of the transform input. Transform-wide invalid signal precedes and succeeds this output and is gated away prior to chirp demodulation.

In order to process contiguous input segments using this configuration, the expander must be impulsed once for each segment. Under these conditions, successive chirps overlap producing undesirable cross-products during chirp modulation and demodulation, and the valid and invalid signal at the output of C1 and C2 merge making invalid-signal removal impossible. Furthermore, unless the valid Fourier transforms are made narrower than the input segments (by specifying chirp filters appropriately), these will also overlap. Some improvement in operation can be gained by making the transform duration appreciably smaller than the input so that there are gaps between valid transforms. With appropriate input filtering, much of the invalid signal would then be confined to these gaps and could be gated away, resulting in less interference. Filtering following chirp demodulation could also improve performance somewhat by suppressing the cross-products.

The problems encountered in the single-channel structure can be avoided using two channels. With only a modest increase in complexity, the single-channel system can be converted into a dual-channel structure by taking full advantage of the facts that most of the time, because of the nature of the signals in the two channels, these signals can be merged without interference and passed through a single string of components, and that the two signals, on an independent basis, can be passed through the same chirp filter in opposite directions (i.e., bilaterally (ref.4)). These techniques make maximal use of components, thereby minimizing size and complexity, while optimizing performance by minimizing the differences between channels.

A dual-channel adaptive filter requires one pair of devices for each chirp filter shown in the single-channel structure of Fig. 1. Two separate devices or a single bilateral device (ref. 4) can be used to realize each chirp-filter pair. These are shown in Fig. 2. With separate devices (Fig. 2a), each channel has its own distinct chirp filter. For a bilateral structure (Figs. 2b and 2c), each channel uses one of two possible

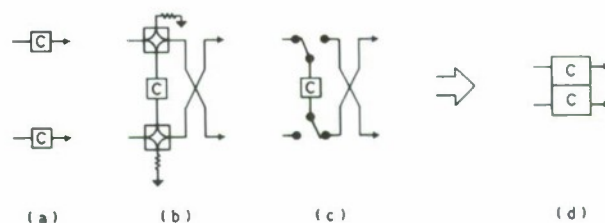


Fig. 2 Chirp-filter channel pairs: (a) two filters; (b) bilateral operation with hybrid networks; (c) bilateral operation with SPDT switches; (d) symbol.

signal-flow directions through a single device, with inputs and outputs timed to prevent coincidence at each port by proper choice of device delay. Compared to separate devices, the bilateral structure requires fewer chirp filters and the channels are better matched, providing greater coherence for overlap summation and better tracking with time and temperature variations, however, since each port is used as both input and output, double-transit and other reflective mechanisms that reflect signal back to the source port become of concern. In general, adequate double-transit suppression can be obtained by making the insertion loss of each chirp filter exceed the expected system dynamic range (measured relative to all spurious and noise). For the wideband CTS adaptive filter, the system dynamic range is limited by the dynamic range of the chirp-generation chain, and is, therefore, maximized by adjusting the expander insertion loss to the point where its double-transit spurious balances the expected thermal-noise level at its output.

The bilateral structure in Fig. 2c uses single-pole-double-throw (SPDT) microwave-diode switches (having independent port controls) to toggle between input and output at each chirp-filter port. This structure is preferable to that of Fig. 2b since the input-to-output isolation of a SPDT switch is typically > 60 dB compared to only about 20 dB for a 3-dB coupler and the loss through a switch is typically < 2 dB while coupler loss is about 3.5 dB. High isolation and low loss are desirable for maximum system dynamic range. Furthermore, high isolation prevents amplifier saturation from feedthrough of high-level input to low-level output.

Figure 3 shows the minimal-component-count dual-channel CTS adaptive filter with signal timing. The combined signals from the two channels are split apart immediately prior to and recombined immediately following each chirp-filter pair. When combined, the signals flow through a single string of amplifiers. The overlapping portions of both the output and the chirps (points 8 and 20, respectively) are demodulated separately from their nonoverlapping portions (points 7 and 19, respectively) and then summed (point 23). This structure maximizes the channel balance at the expense of some imbalance between overlapping and nonoverlapping portions of the signal due to separate amplifier chains at 19 and 20. Maximizing balance when reconstructing the nonoverlapping portions of the signal is justifiable since this portion of the signal generally contains most of the signal energy, particularly when not filtering. The limiter between the C1 and C2 pairs is required to limit anything that extends beyond the level of the MSK spectrum in the current application. The system is thereby optimized for an MSK (noiselike) spectrum, so that power is not wasted in the amplification of undesirable narrowband spectral components. The limiter can also be considered as the system's first stage of narrowband suppression; however, this suppression is highly dependent on signal level. Fourier-transform

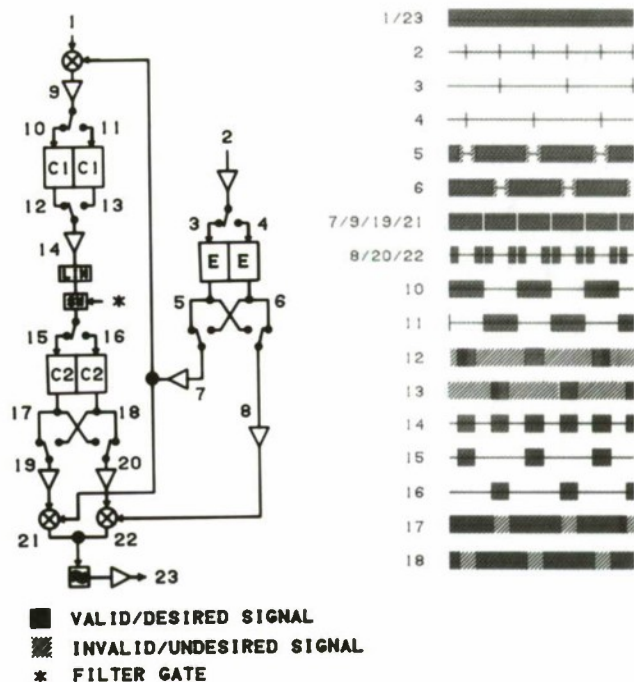


Fig. 3 Dual-channel adaptive filter and signal timing.

notching is provided by the RF switch prior to the C2 pair.

The signal timing shown in Fig. 3 is appropriate for a system structured around bilateral chirp filters. In this respect, the valid portions of the signal pairs 3 and 6, 4 and 5, 10 and 13, 11 and 12, 15 and 18, and 16 and 17 must not coincide. To realize this, the sum of the Fourier-transform and output-segment durations cannot exceed the duration of twice an input segment, while compressor center-frequency group delays must equal this value. This latter requirement is also necessary to insure that chirp demodulation is properly aligned. Impulse-response duration (dispersion) is minimized for each compressor when equal to the sum of the durations of the device's gated input and valid output. Dispersions larger than this are not only unnecessary, but can actually cause some overlap of valid and invalid output. Therefore, using the maximum-allowed transform duration, C2's dispersion must equal twice the input-segment duration, while C1's should be smaller than this by the amount of the output overlap. The minimal dispersion for the expander is equal to the length of an output segment, while its pedestal of delay (delay to start of impulse response) must exceed the duration of an input segment by some reasonable amount.

### III. Performance Simulation

The CTS adaptive filtering technology is intended to be used as a preprocessor for an



MSK-modulated PN-coded signal prior to matched filtering. An FFT-based computer simulation was run to aid in the design by determining ideal performance bounds in the absence of distortions. The simulation processed a signal consisting of a long PN code with an RF interferer. Interference-to-signal (I/S) ratios of 10 dB and 30 dB were used. The combined signal was segmented with and without input weighting, the segments individually Fourier then inverse transformed and the filtered segments recombined with and without overlap. The output signal was then correlated against a reference code and the interference due to the RF signal or its residue was measured relative to the matched-filter correlation-spike amplitude. The simulations were run with and without notching and limiting. An output-segment overlap of 25 %, notch widths from 350 kHz to 550 kHz (6 dB BW, Gaussian) and 30-dB of limiting were used.

The simulation showed that a 350-kHz notch with limiting produces only about 10 dB of interference suppression when no output overlap is used. With output overlap, however, this suppression increases to about 40 dB. By themselves, notching produces about 34 dB, while limiting produces about 26 dB. When the notch width is increased to 550 kHz, the suppression, without limiting, increases to about 56 dB, but limiting degrades this to about 44 dB by disrupting the otherwise smooth shape of the notch. A notch width of about 450 kHz appears optimal for constant suppression independent of limiting. Although suppression for notch widths much greater than 550 kHz is also unaffected by limiting, wide notch widths are undesirable. It was found that thirty 450-kHz notches randomly distributed throughout the MSK spectrum degraded the matched-filter processing gain by only about 1 dB. It was also observed that, without overlap summation, input weighting for sidelobe suppression in the Fourier domain produces no noticeable improvement in suppression capability under all conditions. These results indicate that under ideal (zero-distortion) conditions, greater than 30 dB of narrowband suppression capability is realizable from a CTS adaptive filter employing overlap summation with 25 % overlap, 400 kHz notches and 30 dB of limiting.

#### IV. Specifications and System Performance

The system and device specifications in Table 1 were chosen as an optimal balance between desired system performance and device fabrication complexity for a system bandwidth of 80 MHz. A minimum notch width of 400 kHz and a 25 % fractional output overlap were chosen. Most of the remaining specifications are functions of these few. The device center frequencies were chosen to minimize mixer spurious as well as device percentage bandwidths. The insertion losses and feedthrough/reflection specifications are appropriate for a 38-dB to 40-dB system dynamic range. The expander weighting compensates for the spectral shape of a 4-cycle impulse at 430 MHz. Reflective-array-compressor (RAC) chirp filters have been fabricated to these specifications and tested for performance

#### SYSTEM

BANDWIDTH	B	80 MHz
MINIMUM NOTCH WIDTH	$f_n$	400 kHz
OUTPUT-SEGMENT OVERLAP	$T_0 = 1/f_n$	2.5 $\mu$ s
FRACTIONAL OVERLAP	$T_0 / T$	.25
INPUT-SEGMENT DURATION	$T = 4T_0$	10 $\mu$ s
FT DURATION	$T - T_0$	7.5 $\mu$ s
OUTPUT-SEGMENT DURATION	$T \cdot T_0$	12.5 $\mu$ s
FT RESOLUTION	$1/T$	100 kHz
NOS. OF FT RES. CELLS	TB	800
FT FREQ/TIME (CHIRP SLOPE)	$B/(T - T_0)$	10.7 MHz/ $\mu$ s
SYSTEM DELAY	4T	40 $\mu$ s

#### DEVICE

	E	C1	C2
DISPERSION	12.5 $\mu$ s	17.5 $\mu$ s	20 $\mu$ s
BANDWIDTH	133 MHz	187 MHz	213 MHz
	(= dispersion x chirp slope)		
CENTER FREQUENCY	430 MHz	730 MHz	730 MHz
CT. FREQ. GROUP DELAY	19 $\mu$ s	20 $\mu$ s	20 $\mu$ s
CHIRP TYPE	down	up	down
INSERTION LOSS	40 dB	50 dB	50 dB
WEIGHTING	$x/\sin(x)$	uniform	uniform
FEEDTHRU/REFLECTIONS	<-120 dB	<-90 dB	<-90 dB

Table 1 Adaptive-Filter Specifications.

characterization in the adaptive filtering configuration shown in Fig. 4. Because of a lack of appropriate switches, the less desirable bilateral structure of Fig. 2b had to be used and a configuration in which only the chirp filters

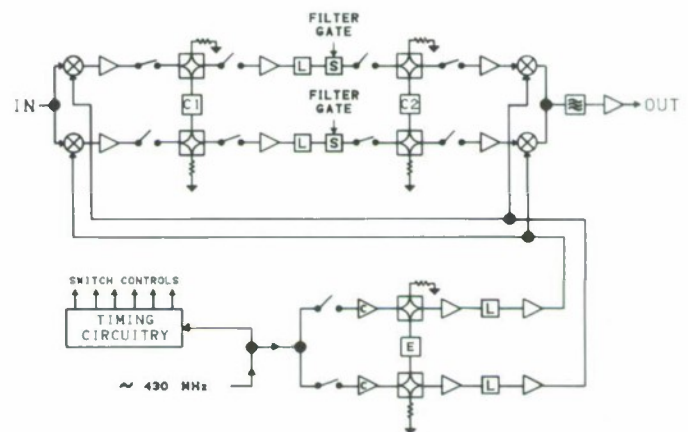


Fig. 4 Demonstration system.

are shared had to be adopted. In this configuration, the switches prior to C1 and C2 have high on/off isolation in order to minimize interference from coupler feedthrough to the low-level outputs. The switches following C1 and C2 have low switching transients and adequate isolation to suppress the high-level coupler feedthrough of each compressor's input (both required to prevent amplifier saturation). Class C amplifiers minimize power and provide additional on/off isolation prior to the expander, while coupler feedthrough of the impulses is suppressed

by following each amplifier in the chains with a limiter. Each channel utilizes its own noncontiguous, nonoverlapping chirp stream equivalent to points 5 and 6 in Fig. 3. All switches are controlled by signals derived by counting down a nominal 430-MHz output from a tunable oscillator, whose output is also used for impulse generation. The system is calibrated by adjusting the frequency of the oscillator until chirp demodulation is properly aligned. This occurs simultaneously in both channels due to an appropriate level of balance between the two.

Figure 5 shows the Fourier transform, in a single channel, for an MSK input in the top trace and the same MSK with a 295-MHz RF interferer, at a 20-dB interference-to-signal ratio, in the bottom trace. Both transforms possess 400-kHz-

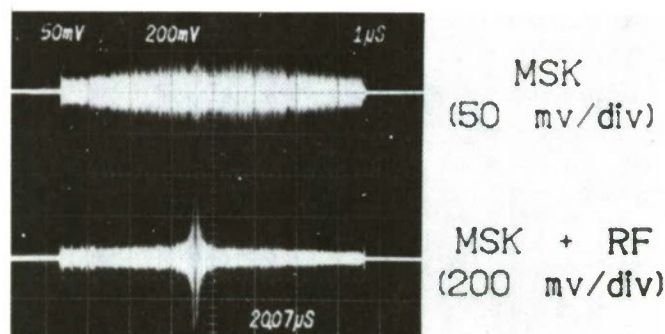


Fig. 5 Notched FT's - single channel.

wide notches at 295 MHz and display the signal spectrum extending from 260 MHz to 340 MHz, left to right. Residual of the spectral peak due to the RF interferer is shown in the bottom trace. This residual produces an output for each segment that is canceled by overlap summation of successive output segments. Figure 6 shows the continuous system output for an MSK input in the top trace and for a 295-MHz RF in the second and third traces. Both signals have been input at the same level. The MSK output (synchronized to the code so that its code structure can be seen) is unaffected by filtering. However, the RF signal (shown synchronized to system timing) has been appreciably suppressed. The untuned output in the second trace shows constructive summation of the overlapping residual RF of consecutive output segments. This disappears in the properly tuned output shown in the bottom trace (the spikes in this trace are due to slight overlap of the input segments). Basic principles indicate that greater than 20 dB of suppression requires phase and amplitude to be controlled through the system to within  $\pm 6^\circ$  and  $\pm 1$  dB, respectively, across the band, while 30 dB requires  $\pm 2^\circ$  and  $\pm .3$  dB, respectively. Demonstration results and phase-and-amplitude measurements indicate a current overall narrowband suppression capability of between 15 dB and 20 dB. It is expected that system-level equalization of amplitude and phase will boost this to the 25 dB to 30 dB range. In the demonstration system the Fourier transforms

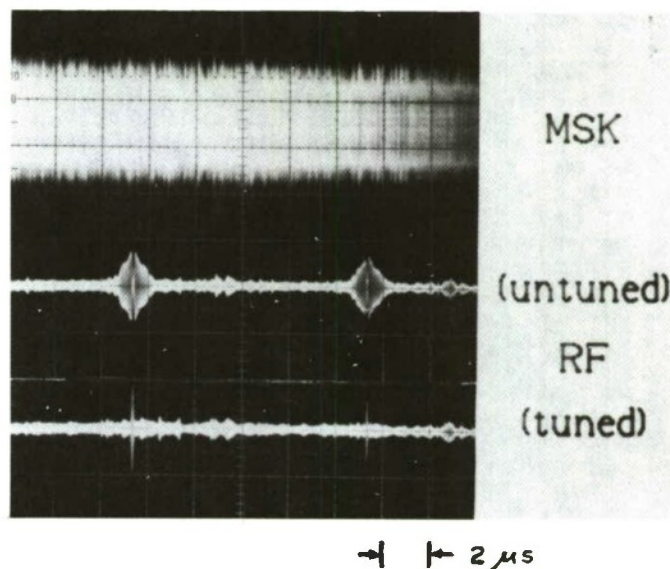


Fig. 6 System outputs for MSK (top trace) and RF (bottom two traces) at equal levels at system input.

were flat to within  $\pm 1.5$  dB, the system delay with the compressors replaced with 50-dB pads was measured at about 60 ns and an expander loss of 47 dB (instead of the desired 40 dB) produced a dynamic range of about 34 dB.

As a further demonstration of interference suppression, an MSK signal was matched filtered with and without RF interference at an input interference-to-signal ratio of 20 dB. The matched-filter processing gain was about 20 dB. The results in Fig. 7 were obtained without CTS adaptive filtering. In the bottom trace the CW interference overwhelms the correlation spike seen without interference in the top trace. The results of Fig. 8 were obtained with CTS adaptive preprocessing of the matched-filter input (see the transforms in Fig. 5). The interference exhibited

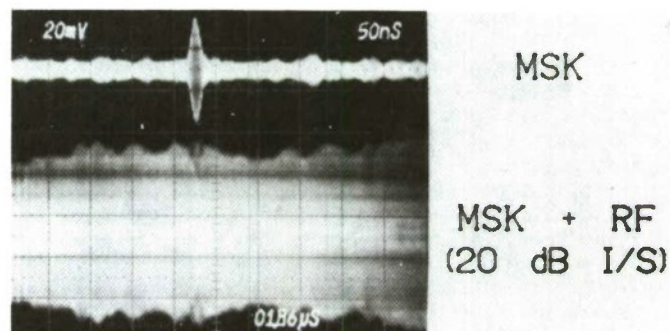


Fig. 7 Matched-filter outputs without adaptive preprocessing: top trace - MSK input only; bottom trace - MSK with added CW interference.





Fig. 8 Matched-filter outputs with adaptive preprocessing: top trace - MSK input only; bottom trace - MSK with added CW interference.

in the bottom trace of Fig. 7 has been appreciably reduced in the bottom trace of Fig. 8 because of filtering.

#### V. Conclusions

A chirp-transform adaptive filtering system technology has been described for narrowband suppression and spectral analysis in the front end of a wideband receiver. The technology currently exhibits a modest 15 dB to 20 dB of narrowband suppression capability; however, projected improvements in system structure and compensation are expected to increase this level to upwards of 30 dB. This system is expected to provide a wideband receiver with the capability to adaptively and simultaneously suppress as many as 30 narrowband interferers with only about 1 dB of implementation loss. Dynamic range is expected to be in the vicinity of 40 dB, which is appropriate for a 30-dB suppression capability. In addition to filtering, the system will provide the valuable function of supplying spectral information to the receiver.

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